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(54) Title: METHOD AND APPARATUS FOR NESTING SECONDARY SIGNALS WITHIN A TELEVISION SIGNAL

(57) Abstract

A method and apparatus (50) is disclosed for encoding secondary signals (54), such as audio signals, within a television signal (52) in a manner such that the signal may be either recorded on a video cassette or broadcast. The audio samples (54) are encoded only on selected lines of each video frame which do not interfere with dedicated broadcast and cable lines and are not affected by head switching of a video cassette recorder. The audio samples (54) are encoded only within an overscan portion of the selected lines such that the encoded samples do not interfere with other portions of the video signal (52). However, by encoding the samples within the overscan portion, the samples do not affect the displayed image. The secondary audio signal (54) is sampled during equally spaced intervals throughout the video frame, then compressed and encoded only within selected horizontal lines. During decoding, the audio samples are decompressed and played back at the original sampling rate to provide audio playback.

METHOD AND APPARATUS
FOR
NESTING SECONDARY SIGNALS WITHIN
A TELEVISION SIGNAL

BACKGROUND OF THE INVENTION

1. Field of the Invention:

The invention generally relates to television
signals and in particular to a method and apparatus for
nesting or encoding secondary signals, such as audio
signals, within a television signal.

2. Description of Related Art:

For certain television applications it is desirable
to nest or encode secondary signals, such as audio
signals, within a television signal. One such
application involves interactive television toys wherein
a teddy bear or other toy receives television signals
provided by a VCR for display on a television set and
extracts and responds to secondary signals nested within
the television signal.

Typically, a control module is interconnected
between the VCR and the television set. Preprogrammed
video cassettes having television programs containing
the nested signals are played using the VCR. The
television signal is intercepted by the control module
and nested audio signals, as well as various nested
control signals, are extracted. The control module
transmits the extracted signals to the teddy bear or
other toy which includes internal mechanisms and

circuitry for movement and speech. The teddy bear
receives the extracted audio signals and plays back the
audio signals while also moving in synchronization.

The entire television signal, including the nested
audio signals, is transmitted from the control module
into the television set which displays the prerecorded
television program. The audio and control signals are
nested or encoded within the television signal in a
manner such that the television does not respond to, and
is not affected by, the presence of the nested signals.
Typically, the audio and control signals nested within
the television signal are synchronized to portions of
the television program such that the teddy bear or other
toy appears to interact with characters and events
displayed on the television set within the television
program.

To implement such a system, the secondary voice and
control signals are encoded within a portion of the
television signal such that the secondary signals can be
extracted from the television signal, yet such that the
television itself does not respond to, and is not
affected by, the presence of the secondary signals. A
number of such techniques have been developed.
Typically, portions of the television signal which do
not provide video, audio or control signals to the
television set are selected and secondary signals are
encoded therein.

One example of such a technique is set forth in
U.S. Patent No. 4,665,431 to Cooper, wherein amplitude
modulated audio samples are encoded on the "back porch"
of a horizontal blank interval of a television signal.
Although the method of Cooper may be effective for
certain applications, it is not particularly effective

for encoding signals within a television program
recorded on a video cassette due to band-width
limitations in consumer grade VCR's.

Another example of a technique for encoding signals
within a television signal is set forth in U.S. Patent
No. 3,857,998 to Justice wherein secondary audio signals
are encoded within a television signal using a chroma
carrier of the television signal. In the technique of
Justice, the secondary audio signals are encoded on a
blanked portion of the back porch of the horizontal
blanking interval. The vertical blanking interval of
the television signal is not encoded. Because all of
the lines of each video frame are not encoded with
secondary audio signals, a problem arises in providing a
continuous output of the secondary audio signal as the
encoded signals are extracted from the television
signal. To overcome this problem, and to provide a
continuous audio output, all audio samples, except for
those which would otherwise be nested within the lines
containing the vertical blanking intervals, are delayed
by a predetermined amount. Those signals which would
otherwise be nested within the vertical blanking
intervals are stored at different locations within the
television signal and are not delayed. During the
decoding of the secondary audio signals, delayed and
non-delayed audio samples are recombined to recreate a
continuous audio signal.

Although the method of Justice is useful for
nesting secondary signals within a television program
for playback on a VCR, it may not be easily applied to
television signals which are broadcast because the
secondary signal is stored at locations within the
television signal which contains dedicated broadcast-
related control signals during broadcast. Thus, the

technique of Justice cannot be easily applied to
providing nested signals for controlling the teddy bear
or toys of the type described above wherein the
television program is broadcast rather than stored on a
video cassette. As can be appreciated, there is
considerable advantage to providing such programs via
broadcast television rather than on prerecorded video
cassettes, with the greatest advantage perhaps being the
diversity and variety of programming material that can
be conveniently provided via broadcast television.

     Another technique for encoded audio signals is
described in U.S. Patent No. 4,786,967 to Smith. In the
technique of Smith, as with the technique of Justice,
secondary audio samples are not stored within the
vertical blanking lines. However, unlike Justice, the
audio samples which would otherwise be stored within
those vertical blanking lines are not stored elsewhere.
A discontinuity results from the lack of audio samples
stored in the vertical blanking signals. The
discontinuity results in a 60-cycle hum during playback.
Furthermore, higher-order harmonics based on the 60-
cycle hum may also be present which further undesirably
distort the output audio signal. With the technique of
Smith, the 60-cycle hum problem is addressed by tapering
the encoded audio signal to silence adjacent to the
vertical blanking interval. In other words, rather than
storing secondary audio samples at their full value
throughout the horizontal portion of the signal, the
magnitude of the audio signals are tapered at opposing
ends of the horizontal portion such that the secondary
audio signal is reduced to silence at the transition
times from the display portion to the vertical blanking
intervals. By tapering the audio signal, the 60-cycle
hum generated during playback is reduced. Furthermore,
the higher-order harmonics resulting from a sharp

transition from no audio signal to a full audio signal
are likewise reduced.

Although the technique of Smith may be effective in
eliminating the 60-cycle hum problem without requiring
the complicated delaying and recombination techniques of
Justice, the technique of Smith may not be satisfactory
for broadcast television signals. One reason that the
technique of Smith may not be satisfactory for broadcast
signals is that network and cable broadcasters impose
limitations and constraints which prevent the encoding
of audio signals within all of the video lines of the
horizontal portion of the television signal. In
particular, twenty-two lines of the video signal cannot
be encoded due to such restrictions. The twenty-two
lines which cannot be encoded represent over eight
percent of the overall video signal. The lack of an
audio signal within those twenty-two lines results in an
audible 60-cycle hum and may further result in higher-
order harmonics as well. Moreover, even if the
technique of Smith were modified to also provide
tapering in the vicinity of the twenty-two restricted
lines, such a tapering would, at best, only be partially
successful in eliminating the 60-cycle hum and higher-
order harmonics.

Accordingly, it would be desirable to provide a
technique for nesting secondary audio signals, as well
as other secondary signals, within a television signal
in a manner such that the encoded television signal may
be broadcast, rather than merely provided by a video
cassette recorder, and wherein 60-cycle hum and higher-
order harmonic problems described above are
substantially eliminated. The ability to provide
broadcast television signals with nested audio and
control signals is particularly desirable as a far

greater variety of such programs can be provided conveniently to the customer than can be provided on video cassettes.

Although the audio nesting technique has been described above primarily with reference to the application of an interactive teddy bear or toy, encoded audio and control signals may be employed for a wide range of applications for controlling a wide range of devices, and are not limited to merely toys or other entertainment devices.

SUMMARY OF THE INVENTION

The invention relates to a method and apparatus for encoding or nesting secondary signals, such as audio signals, within a television signal, wherein the television signal may be either broadcast or stored on video cassettes and wherein 60-cycle hum problems associated with prior art techniques are avoided. In accordance with one aspect of the invention, the secondary signals are encoded using amplitude modulation within a non-visible overscan portion of a television signal. The non-visible overscan portion of the television signal represents a small portion of the otherwise viewable portion of the television signal which is effectively clipped from the peripheral portions of a television screen during display. More specifically, a standard television signal provides a generally rectangular video display portion. However, conventional television sets do not display the entire rectangular video portion. Rather, conventional television sets display most of the rectangular display portion but clip portions of the display on the left, right, top and bottom of the rectangular display area.

The portion which is not displayed is generally referred
to as the "overscan" portion. The overscan portion is
provided so that a viewer sees a complete visible image
and does not see any black blanking bars along the edges
of the screen regardless of the electrical
characteristics of a particular television set.

By encoding or nesting the secondary signals, such
as audio signals, within the overscan portion, the
secondary signals do not affect the visible image. The
encoded secondary signals only affect the visible image
if the television is sufficiently out of alignment to
display a portion of the overscan region, and even then
the presence of the secondary signals results only in a
somewhat fuzzy or distorted image on the extreme edge of
the television screen. Yet, by encoding the secondary
signals within the overscan portion, it is assured that
the secondary signals do not corrupt any portion of the
television signal used by network or cable broadcasters.
Hence, a television signal having secondary audio
signals encoded within the overscan portion can be
safely broadcast, and is therefore not limited only to
playback from a prerecorded video cassette.

Thus, the secondary signals are stored within
overscan portions of the visible signal lines. No
secondary signals are stored within the vertical
blanking lines. Furthermore, no secondary signals are
encoded within the lines typically restricted by network
and cable broadcasters. To prevent any 60-cycle hum or
other sound distortion from occurring as a result of the
skipped lines, the secondary audio signal is first
sampled at equally spaced intervals throughout a
corresponding video frame at a rate corresponding to the
number of lines to be encoded within the frame. Then,
the audio samples are encoded in only the non-skipped

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lines. Thus, the audio signal corresponding to the time
period of an entire frame of video is compressed into
only a portion of the lines of the video image. During
decoding, the audio samples are extracted from those
lines then decompressed for audio playback. In this
manner secondary audio samples are provided during
playback at equal time intervals throughout the display
of the corresponding video frame. Accordingly, no
substantial gaps exist between the audio samples and no
60-cycle hum occurs.

In one exemplary embodiment, the foregoing is
achieved by providing a method for encoding secondary
information within a television signal having overscan
portion. The method comprises the steps of selecting
horizontal lines to be encoded with the secondary
information, sampling the secondary information at a
rate related to the number of horizontal lines to be
encoded, and encoding the sampled audio information in
the selected horizontal lines of the television signal
with the overscan portion thereof. The audio
information is encoded by amplitude modulation. Other
exemplary embodiments of the invention are described
herein.

By encoding secondary signals within only the
overscan portion of selected lines of the horizontal
signal, the objectives set forth above are achieved. In
particular, the television signal having the encoded
secondary signals is compatible with cable and broadcast
systems. Moreover, by uniformly sampling the secondary
signals, then compressing the signals during encoding
and decompressing the signals during decoding, the 60-
cycle hum problems described above are avoided. Other
advantages and applications of the invention will be
described below or will be apparent therefrom.

Furthermore, although principles of the invention are
advantageously employed for storing secondary signals,
such as audio signals, within a television signal,
principles of the invention may be employed for other
applications and in other systems.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a graphic illustrating a complete
frame of a television signal, particularly identifying
visible and non-visible portions of the television
signal.

Figure 2 is a graph illustrating a portion of a
single line of a television signal encoded with a
secondary audio signal.

Figure 3 is a sequence of graphs illustrating the
sampling and compression of audio signals in accordance
with the invention.

Figure 4 is a block diagram, partially in
schematic form, illustrating an apparatus for encoding
audio signals onto a television signal.

Figure 5 is a graph illustrating a portion of a
single line of a video signal, also illustrating video
gate and sample gate signals employed during encoding.

Figure 6 is a block diagram, partially in
schematic form, illustrating an apparatus for decoding
television signals encoded by the apparatus of
Figure 4.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS
OF THE INVENTION

Referring to **Figures 1-6**, exemplary embodiments of the invention will now be described. **Figure 1** is a graphical illustration of a single frame 10 of a television signal. Frame 10 includes a video image portion 12, a vertical blanking portion 14 and a horizontal blanking portion 16. The vertical blanking portion and horizontal blanking portion represent signals which are not displayed. Video image portion 12 itself includes a visible image 18 and an overscan portion 20. Visible image 18 represents a portion of the video image which is actually displayed on a properly tuned and calibrated television screen. Overscan portion 20 represents a portion of the video image which is not ordinarily displayed on a properly calibrated television set. The actual size of the overscan portion typically varies among different television sets. However, for conventional television sets, at least a minimal overscan portion is always provided.

In accordance with one aspect of the invention, secondary audio signals 22 are encoded within overscan portion 20. In particular, the secondary audio signals are encoded within the overscan portion between the horizontal blanking portion and the visible image portion of the video frame. By encoding the secondary audio samples within the overscan portion, the samples do not interfere with or corrupt control portions of the vertical or horizontal intervals. Furthermore, the secondary audio signals do not affect the actual visible image, except for television sets which are poorly tuned or calibrated such that a portion of the overscan is displayed. In such cases, some distortion or blurring

of the image on the left hand portion of the television
screen may be apparent.

As will be described in detail below, the secondary
audio samples are compressed during encoding and
decompressed during decoding to provide equally spaced
audio samples throughout the television frame such that
no 60-cycle hum or related distortion occurs. Also, it
should be noted that in a preferred embodiment, the
secondary audio samples are not stored within the
overscan portion of all horizontal lines of the video
image. Rather the secondary audio samples are encoded
only in lines which are not subject to network or cable
restrictions and which do not coincide with head
switching or vertical synchronization locations. Also,
it should be noted that **Figure 1** merely provides a
simplified graphical illustration of a television signal
frame. An actual composite television signal for
providing an entire television image frame is typically
split into odd and even fields with the lines of the
fields provided with control and synchronization
portions which are not specifically illustrated.
Details of actual composite video signals will be
described below or will be apparent to those skilled in
the art.

Figure 2 illustrates an exemplary horizontal line
24 having a secondary audio sample 26. Horizontal line
24 is graphically illustrated as voltage, on the IRE
(Institute of Ratio Engineers) scale, as a function of
time. Line 24 includes a horizontal sync portion 28 and
an active video portion 30. Horizontal sync portion 28
includes a horizontal blanking interval 32 having a
voltage of -40 IRE. Horizontal sync portion 28 also
includes synchronization signals generally represented
by 34. Active video portion 30 includes a visible video

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portion 36 and left and right overscan portions 38 and
40. Secondary audio sample 26 is encoded within left
overscan portion 38. The entire horizontal line
illustrated in **Figure 2** has a "length" of 63.5
microseconds (μs). Each overscan portion has a length
of about 2 μs with the secondary audio sample encoded
within the first 1.2 μs of overscan portion 38.

The amplitude of signal 24 within sample 26 is
encoded to represent the amplitude of a single sample of
the secondary audio signal being encoded. In one
embodiment the amplitude of the secondary audio sample
varies from 7.5 - 100 IRE. However, alternative
amplitude scales may be employed. Furthermore, in the
example illustrated in **Figure 2** only a single sample is
encoded within overscan portion 48. In alternative
embodiments, two or more audio sample points could be
encoded therein. In other alternative embodiments, the
trailing overscan portion 48 could also be employed for
encoding secondary audio samples. However, the
embodiment having only a single audio sample within just
the leading overscan portion is desirable as such allows
a comparatively simple apparatus to perform the encoding
and decoding of the audio samples. The use of two or
more samples per overscan portion or the use of both the
leading and trailing overscan portions could require a
more complicated encoding and decoding apparatus,
although such systems would allow for a higher
resolution of the secondary audio signal.

In the exemplary embodiment thus far described, the
secondary signals have been described as being secondary
audio signals. In alternative implementations, the
secondary signals may provide other information rather
than audio information, and may, for example provide

only control signals for controlling a device receiving
and decoding the secondary signals.

Thus, **Figure 2** illustrates an exemplary horizontal
line having an encoded secondary data sample. As noted,
not all horizontal lines are provided with data samples.
In a preferred embodiment, only lines 22 - 257 and 285 -
520 are provided with secondary signal samples. Lines 1
- 9 and 263 - 271 provide vertical synchronization
signals which may be corrupted by the presence of the
secondary signal. Lines 10 - 21 and 272 - 284 are
employed by broadcast and cable operators as dedicated
non-program lines for encoding control signals and the
like. The presence of secondary signal samples within
the dedicated lines could result in corruption of the
dedicated signals provided therein. Lines 258 - 262 and
520 - 525 are typically employed as head-switching
locations within VCR's. Accordingly, secondary signal
samples encoded therein might not be properly decoded
and indeed may hinder proper head switching. Thus, only
lines 22 - 257 and 285 - 520 are preferably employed for
storing secondary signal samples. Of these lines, lines
22 and 23 and 285 and 286 are employed for encoding
control bits, described below. Accordingly, lines 24 -
257 and lines 287 - 520, totaling 464 lines, are used
for encoding audio signals samples.

Thus, only selected lines are employed for storing
audio samples. If secondary audio samples were merely
stored within those lines without time compression and
decoded without decompression, a resulting secondary
audio signal would likely have a 60-cycle hum resulting
from gaps between audio samples as a result of the
skipped lines. To avoid the 60-cycle hum, the method
and apparatus of the invention compresses equally spaced
audio samples during encoding and decompresses the audio

samples during decoding to provide a continuous audio
output unaffected by discontinuities.

Figures 3A - 3F illustrate an exemplary audio
signal 42 in various stages of processing. More
specifically, **Figure 3A** illustrates an input analog
secondary audio signal 42. **Figure 3B** illustrates
sampling points for sampling signal 42. A total of 24
sampling points are illustrated. **Figure 3C** illustrates
a resulting sampled digital signal 44. **Figure 3D**
illustrates usable 45 and unusable 47 portions of the
corresponding video signal. The unusable portion
represents, for example, head switching lines. To avoid
the unusable portion, the digital audio signal is
compressed within only the usable portions, as
illustrated in **Figure 3E**. No samples are lost or
discarded. Rather, the time spacing between the samples
is compressed to allow the entire number of original
samples to be stored within only the usable portions of
the video signal. A similar sequence of steps are
performed during decoding to decompress signal 46 to
yield a continuous digital representation of input
analog signal 42.

It should be noted that **Figures 3A - 3F**
illustrate only an exemplary audio signal having 24
digital sampling points and an exemplary video signal
having only a single unusable portion. In the preferred
implementation, as many as 464 audio samples may be
stored in a manner which avoids several unusable
portions of the television signal.

Figure 4 is a block diagram of an encoder 50
employed for receiving a television signal and a
secondary audio signal and for compressing and encoding
the secondary audio signal into the television signal.

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Encoder 50 receives a television signal, in NSTC format, through video input terminal 52 and a secondary audio signal through audio input terminal 54. Encoder 50 outputs the television signal with encoded secondary audio samples through output terminal 56. Encoder 50 may also receive a switched input signal through input terminal 58 and a manual gate control signal from manual switch 60. The switch signal received through terminal 58 and a manual gate control signal provided by switch 60 are provided for defining control bits for encoding within the output television signal which define whether corresponding horizontal lines contain valid audio samples. Such will be described in detail below.

The video input signal received through terminal 52 is processed by a synchronization separation and DC restore block 62 which outputs a composite synchronization signal to a timing and control block 64. Block 62 also outputs the video signal along line 66 to a video gate 68 for further transmission through an output amplifier 70 onto video output terminal 56. Video gate 68 is open at all times other than when secondary signals are encoded onto the video signal. The video signal is deactivated for a period of time while secondary signals are added, thus preventing a super-position of the video signal and the secondary signal.

Video gate 68 is controlled by a video gate control signal 72 provided by timing and control block 64. **Figure 5** illustrates video gate control signal 72. **Figure 5** also illustrates a sample gate signal 74 which controls an audio gate 76 (**Figure 4**) which provides sampled audio signals. As can be seen from **Figure 5**, the video gate and sample gate signals are mutually exclusive, with the video gate signal being off only

while the sample gate signal is on. In this manner,
audio samples are encoded only during a time period when
the video signal is disconnected from the output.
Figure 5 also illustrates an exemplary portion of the
video output signal 76 having an encoded audio sample
78.

Returning to **Figure 4**, the manner by which the
audio signals are sampled and the video gate and sample
gate signals are generated will now be described. As
noted, timing and control block 64 receives the
composite synchronization signal. Timing and control
block 64 is preprogrammed or pre-configured with
information representing the number of audio samples to
be encoded within each frame of the output video signal.
Timing and control block 64 employs that information to
output an audio sampling pulse along line 78 to control
a sample and hold gate 80 for sampling the audio signal
received from terminal 54 through amplifier 82. As an
example, if encoder 50 is configured to encode 232 audio
samples within a field displayed for 1/60th of a second,
timing and control block 64 outputs pulses at a rate of
232 divided by 60 or 71.8 microseconds. Hence, sample
and hold gate 80 is periodically opened every 71.8
microseconds to sample the audio input signal. The
sampled audio signals received through gate 80 are
converted to digital bytes by an analog-to-digital (A/D)
converter 82 for storage within a memory 84. In an
exemplary implementation, memory 84 is capable of
storing 232 bytes, although other memory sizes may be
employed. In this manner, 232 equally spaced audio
samples are detected and stored in memory 84. The audio
samples are equally spaced throughout the entire time
period defined by the first field of the input video
frame. This is in contrast to some prior art systems
which sample a secondary audio signal only during

portions of each video frame resulting in gaps or
discontinuities in an output audio signal and thereby
resulting in sound distortion including the above-
described 60-cycle hum.

Thus, the secondary audio signal samples are
converted to digital format and stored in memory 84. It
should be noted that encoder 50 begins sampling the
audio signal immediately within the video frame, even
though the audio samples are not encoded into the video
frame until after at least the twenty-second horizontal
line is received.

The encoding of the stored audio samples onto the
output video signal is controlled by timing and control
block 64. Initially, timing and control block 64
locates the horizontal lines wherein the audio samples
are to be encoded. More specifically, timing and
control block 64 locates line 24 and outputs appropriate
sample gate and video gate control signals during the
overscan portion of line 24 to encode of an audio sample
therein. Simultaneously, timing and control block 64
outputs a read signal to memory 84 for outputting the
first audio sample along a data bus to a D/A converter
86 for subsequent transmission of the analog audio
sample through audio gate 76 and onto the video signal
output through video out 56. Timing and control block
64 also outputs the memory address corresponding to the
first analog sample to memory 84. In practice, timing
and control block 64 may need to transmit the memory
address and read signal to memory 84 somewhat in advance
of the overscan period to allow time for the audio
sample to be output from memory 84, converted to analog
by D/A converter 86, and applied to audio gate 76.

In this manner, the first audio sample is encoded on the output video signal during the overscan period of the 24th horizontal line. To encode the remaining samples of the first field, timing and control block 64 periodically outputs additionally read signals and corresponding memory address signals to memory 84 while also controlling the video gate and sample gate signals to encode a total of 232 audio samples with horizontal lines beginning at line 24. To encode the remaining 232 samples within the second field, timing and control block 64 locates horizontal line number 287 and provides the appropriate read, memory address signals, video gate and sample gate signals for encoding the remaining 232 audio samples on consecutive horizontal lines beginning with line 287.

Thus, a total of 464 secondary audio samples are encoded onto the output video signal within each frame of the video signal. If desired, encoder 50 can be configured to encode more or fewer audio samples.

Thus, timing and control block 64 operates to locate horizontal lines 24 and 287 for use in encoding audio sample beginning at those lines. Timing and control block 64 also locates other lines within the video frame for use in encoding control bits. In particular, control bits are encoded at lines 22 and 23 and 285 and 286. Horizontal lines 22 and 285 receive a manually setable control bit which indicates the presence of valid audio samples. The value of the manually setable bit is controlled by setting manual gate control 60 to either an on position, as shown, or to an off position. In the on position, manual gate control 60 is connected to a source which provides a 7.5 IRE signal to a first control bit data gate 88, also labeled data 1 gate. Gate 88 receives a signal from

timing and control block 64 only during the appropriate
overscan period of the 22nd and 285th horizontal lines.
Thus, when the manual gate control is set to the on
position, a control bit level of 7.5 IRE is encoded
within the overscan portion of horizontal lines 22 and
285. As will be described below, a decoder (which
receives the encoded video signal) detects the signal
value within lines 22 and 285 and plays back the sampled
audio signals only if an IRE value of 7.5 is encoded
therein. To indicate invalid audio sample data, manual
gate control 60 is manually switched to an off position
which provides a signal of 100 IRE to gate 88 for
encoding onto lines 22 and 285. The signal value level
of 100 IRE provides an indication to the decoder that
the secondary audio signals are invalid. Accordingly,
the decoder ignores any audio samples encoded within the
overscan portions. In this manner, the playback of
secondary audio signals can be manually disabled during
encoding. The manual gate control is advantageously
employed to ensure that secondary audio signals are not
extracted from the video signal, for example, during the
broadcast of commercials rather than program material.

A second control bit is provided within lines 23
and 286. The second control bit is applied to the video
signal through a second control gate 90 which receives
enabling signals from timing and control block 64 only
during selected overscan portions of horizontal lines 23
and 286. Gate 90, also labeled data 2 gate within
Figure 4, receives input signals along line 92 from a
comparitor 94 which has a first input connected through
a switch 96 to either amplifier 82 or switched signal
input terminal 58. With switch 96 set to an internal
mode, an audio signal output by amplifier 82 is applied
to the first input of comparitor 94. A second input of
comparitor 94 is connected along a line 98

interconnecting a voltage source 100 and a ground 102
through a variable resistor 104. The variable resistor
104 is used to select the level of audio input that
causes comparitor 94 to go high with an audio signal
input thereto. In this manner, an audio threshold can
be set to a level above any inherent noise level.

        As interconnected, comparitor 94 provides an active
high signal if an audio signal is present and if switch
96 is set to the internal mode. Comparitor 94 provides
an inactive low signal if no audio signal is present
while switch 96 is connected to the internal mode.
Hence, a control bit is encoded within the 23rd and
286th lines which indicate whether an audio signal is
present. Thus, if no secondary audio signal is input,
the lack of the audio signal is represented by the
second control bit. The decoder (not shown in **Figure
4**) examines the second control bit prior to decoding the
samples stored within the overscan portions of the
horizontal lines.

        With switch 96 set to the external position, the
value of the second control bit is controlled by a input
switch signal received by terminal 58. The external
position provides for leading and lagging of the active
audio signal. More specifically, a control signal
indicating the presence of audio samples may be set
within a video field which is several fields in advance
of an initial field containing actual audio samples.
This function allows the decoder, described below, to
turn itself on slightly before a valid audio signal is
present to thereby allow a "soft" turn on of the audio
output of the decoder. The soft turn on eliminates gate
switching noise which may occur if the decoder is
activated or deactivated while actual secondary audio
signals are present. Hence, by setting switch 96 to the

external position at appropriate time, the operation of
the decoder may be activated slightly before the
presence of audio samples and deactivated slightly after
the completion of audio samples. This helps to avoid
any audible "pop" within the audio signal output by the
decoder.

Thus, lines 22, 23, 285 and 286 receive control
bits which identify the validity of audio samples
encoded. In the foregoing, for clarity, the encoding of
the audio samples was described before the encoding of
the control bits. In operation, the control bits of
lines 22 and 23 are encoded prior to the audio samples
beginning at line 24. Likewise, the control bits of
lines 285 and 286 are encoded prior to the audio samples
beginning at line 287.

The above-described process for encoding control
bits and audio samples within a single video frame is
repeated for each consecutive video frame, until, for
example, an entire television program is encoded with
audio signals, perhaps for use with the interactive
teddy bear or toys of the type described above.

Referring to **Figure 6**, the operation of a decoder
110 for extracting the control bits and audio samples
from an encoded video signal will now be described.
Decoder 110 receives the encoded television signal
through a video input terminal 112. The video input
signal is applied through a capacitor 114 to a
synchronization separator and DC restore block 116.
Block 116 outputs a composite synchronization signal to
a timing and control block 118 and also outputs a
clamped video signal to a sample and hold gate 120.
Sample and hold gate 120 is controlled by timing and
control block 118 to transmit portions of the video

input signal only during the predetermined overscan portion of the selected horizontal lines. More specifically, timing and control block 118 locates line 22 within the video input signal and transmits an enabling signal to gate 120 to allow the control bit provided within line 22 to be passed therethrough. Sample and hold gate 120 thereby outputs a signal for charging sample and hold capacitor 122 to a value representative of the encoded pulse on line 22. The resulting signal is applied to a high comparison block 124 where it is latched therein. High compare block 124 also receives a latch enable signal from timing and control block 118. Next, timing and control block 118 locates line 23 and applies a signal to gate 120 causing the control bit encoded therein to be applied to low compare block 126. Low compare block 126 also receives a latch enable signal from timing control block 118. The outputs of the high and low compare blocks are then applied to an AND gate 128, the output of which is applied to an audio gate 130 for activating audio output. In this manner, audio gate 130 is opened only if the control bits of lines 22 and 23 are both high. A low bit in either or both of lines 22 and 23 prevents the opening of audio gate 130 and thereby prevents the secondary audio signals from being played back.

Timing and control block 118 also locates line 24 and applies a signal to gate 120 for transmitting an audio sample pulse therethrough to an A/D converter 132. The audio sample of line 24 is digitized then stored within a twenty-eight line ring buffer 134. Immediately upon sampling line 24, timing and control block 118 also begins outputting data from buffer 134 in first-in/first-out (FIFO) fashion. Data output from buffer 134 is identified by an address provided by timing and control block 118 through address counter 136. The

bytes output from buffer 134 are transmitted to a D/A
converter 138. A resulting analog audio signal is
transmitted through audio gate 130 to an output
amplifier 140 for playback on a speaker 142. As noted
above, audio gate 130 is only open if high signal values
are encoded within the control bits of both lines 22 and
23.

The process of sampling the video input signal,
storing the samples within ring buffer 134, and
outputting data from ring buffer 134 proceeds throughout
the first field of the input video frame. The process
repeats for the second field with timing and control
block 118 locating lines 185, 186 and 187 and applying
appropriate control signals to gate 120 and to high and
low compare blocks 124 and 126.

It should be noted that the timing for reading from
buffer 134 and writing to buffer 134 is asynchronous.
The write timing is based on a horizontal
synchronization timing of 63.5 microseconds. The read
timing is based on the audio sampling timing of 71.8
microseconds. Based this timing, when all 232 samples
of a single field have been digitized into memory, 27
bytes of data remain to be clocked out over the
remaining 30 scan lines. In this manner, the ring
buffer is completely empty when the first audio byte of
the next field is ready to be sampled.

By employing a ring buffer, the audio samples which
are compressed within only the selected lines of the
input video frame are decompressed during playback.
Hence, the audio playback provides equally spaced audio
samples through the video frame, rather than compressed
audio samples during only a portion of the video frame.
As noted, by providing equally spaced audio playback

samples, audio distortion such as 60-cycle hum is
eliminated.

Thus, all 464 audio samples corresponding to an
entire video frame are decompressed and played back
during a time period equal to the length of the video
frame. It should be noted, however, that the playback
of the first audio sample does not begin until after
timing and control block 118 identifies line 24. As
such, the output of the audio samples is delayed or
skewed from the input video signal by at least a time
period equal to the first 23 lines of the video frame.
To compensate for this encoding delay or skew, the
synchronization of the audio source can be advanced by
28 lines during encoding to achieve exact alignment.
However, the synchronization skew is very small (about
1.78 μs) and therefore can be ignored for most
applications.

Appendix A provides circuit schematics for an
exemplary embodiment of the encoded and decoder.

In the exemplary system thus far described, control
bits are employed only to indicate the presence or
absence of valid audio signals. Additional control
signals may be employed, for example, to select between
two tracks of audio signals, perhaps corresponding to
two characters within a television program. As another
example, further control bits can be employed for
identifying the location of the audio samples within the
video signal. For example, a control bit may indicate
whether the samples are stored within the leading
overscan portion or the trailing overscan portion of
each horizontal line. As can be appreciated, a wide
range of features may be exploited with the appropriate
choise of control bits and encoding schemes.

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Furthermore, the system need not be employed only for
encoding audio signals but may encode any form of
secondary information. Including, for example, textural
information encoded and digital format.

What has been described is a method and apparatus
for encoding secondary signals, such as audio signals,
within a television signal in a manner such that the
signal may be either recorded on a video cassette or
broadcast. The audio samples are encoded only on
selected lines which do not interfere with dedicated
broadcast and cable lines and are not affected by head
switching of a video cassette recorder. The secondary
audio signal is sampled during equally spaced intervals
throughout a video frame, then compressed and encoded
only within selected horizontal lines. During decoding,
the audio samples are decompressed and played back at
the original sampling rate to provide an audio playback
which is unaffected by the audio distortion which can
occur in prior art techniques as a result of gaps or
discontinuities within the audio signal. Although the
invention has been primarily described with respect to
the encoding and decoding of a secondary audio signal,
principles of the invention may be employed for encoding
or decoding any type of information within an NTSC
television signal and may be further applied to other
types of signals as well such as PAL and Secan, and to
non-television signals. Likewise, although the
invention is advantageously employed for encoding audio
signals for use with an interactive teddy bear or toy, a
wide range of other applications may be employed.
Hence, the exemplary embodiments described herein are
merely illustrative of principles of the invention and
should not be taken as limiting the scope of the
invention.

CLAIMS

1.   A method of transmitting secondary information
within a television signal, said television signal
having a first plurality of lines, said method
comprising the steps of:
        selecting a second plurality of lines among said
first plurality of lines;
        generating a plurality of samples from said
secondary information at a compressed rate;
        encoding said plurality of samples within said
second plurality of lines; and
        transmitting said television signal containing said
secondary information.

2.   The method of claim 1, wherein said secondary
information is encoded by amplitude modulation.

3.   The method of claim 1, wherein the secondary
information includes audio information.

4.   The method of claim 1, wherein the secondary
information includes control information.

5.   The method of claim 4, wherein the control
information identifies valid portions of secondary
information.

6.   The method of claim 1, wherein the step of
generating said plurality of samples at said compressed
rate includes the step of compressing said plurality of
samples within said second plurality of lines; and

wherein said second plurality of lines excludes
lines employed by broadcast and cable operators or as
head switching locations.

7.   The method of claim 1, wherein when said
plurality of samples are generated at said compressed
rate, a time spacing between said plurality of samples
is compressed to encode all of said plurality of samples
within said second plurality of lines so that said
plurality of samples are stored within only usable
portions of said television signal.

8.   The method of claim 1 further comprising:
receiving said television signal containing said
secondary information;
    extracting said plurality of samples from said
second plurality of lines; and
    decompressing said plurality of samples.

9.   The method of claim 8, wherein the step of
decompressing said plurality of samples includes the
step of outputting said plurality of samples at
substantially equally spaced time intervals.

10.   The method of claim 1, wherein each of said
second plurality of lines has an active video region,
and each of said active video regions includes an
overscan portion; and
    wherein the step of encoding said plurality of
samples includes the step of encoding said plurality of
samples within said overscan portions of said second
plurality of lines.

11.   A method of decoding secondary information
encoded within a television signal, said television
signal having a first plurality of lines, a plurality of
samples generated from said secondary information, said
method comprising the steps of:
      receiving said television signal containing said
secondary information;
      selecting a second plurality of lines among said
first plurality of lines;
      extracting said plurality of samples from said
second plurality of lines; and
      decompressing said plurality of samples.

12.   The method of claim 11, wherein the step of
decompressing said plurality of samples includes the
step of outputting said plurality of samples at
substantially equally spaced time intervals;
      wherein each of said second plurality of lines has
an active video region, and each of said active video
regions includes an overscan portion;
      wherein the step of extracting said plurality of
samples includes the step of extracting said plurality
of samples from said overscan portions of said second
plurality of lines; and
      wherein said second plurality of lines includes
usable lines and not unusable lines.

13.   An apparatus for encoding secondary
information within a television signal, said television
signal having a first plurality of lines, said apparatus
comprising:

a first terminal for receiving secondary
information;

a second terminal for receiving a television
signal;

means for generating a plurality of samples from
said secondary information at a compressed rate;

means for selecting a second plurality of lines
among said first plurality of lines; and

means for encoding said plurality of samples within
said second plurality of lines.

14.   The apparatus of claim 13, wherein said means
for generating said plurality of samples at said
compressed rate includes means for compressing said
plurality of samples within said second plurality of
lines; and

wherein said second plurality of lines excludes
lines employed by broadcast and cable operators or as
head-switching locations.

15.   The apparatus of claim 13, wherein when said
plurality of samples are generated at said compressed
rate, each of said plurality of samples is associated
with its corresponding one of said second plurality of
lines;

wherein each of said second plurality of lines has
an active video region, and each of said active video
regions includes an overscan portion; and

wherein said plurality of samples are encoded with
said overscan portions of said second plurality of
lines.

16. The apparatus of claim 13 further comprising:

a first circuit for providing timing and control signals;

a synchronization separation and DC restoration circuit coupled to said second terminal and said first circuit for outputting a composite synchronization signal to said first circuit and a video signal;

an amplifier coupled to said first terminal for amplifying said secondary information;

a sample and hold gate controlled by said first circuit for passing a sample of said amplified secondary information;

a sample and hold circuit coupled to said sample and hold gate for holding said sample of said amplified secondary information;

an analog-to-digital converter coupled to said sample and hold circuit for converting said sample into a digital sample, said analog-to-digital converter controlled by said first circuit;

a memory coupled to said analog-to-digital converter for storing said digital sample, said memory controlled by said first circuit;

a digital-to-analog converter coupled to said memory for converting said digital sample into an analog sample, said digital-to-analog converter controlled by said first circuit;

a video gate coupled to said synchronization separation and DC restoration circuit for passing said video signal, said video gate controlled by said first circuit; and

an audio gate coupled to said digital-to-analog converter for passing said analog sample, said audio gate controlled by said first circuit,

wherein when said video gate is on, said audio gate is off, and when said audio gate is on, said video gate is off;

wherein said means for generating said
plurality of samples from said secondary information at
said compressed rate includes said first circuit, said
sample and hold gate and said sample and hold circuit;

wherein said means for selecting said second
plurality of lines among said first plurality of lines
includes said first circuit; and

wherein said means for encoding said plurality
of samples within said overscan portions of said second
plurality of lines includes said first circuit, said
analog-to-digital and digital-to-analog converters, said
memory, and said video and audio gates.

17.   The apparatus for claim 16 further comprising:
a first data gate for passing a first signal
indicating that said secondary information is valid;

a comparator coupled to said amplifier for
comparing said amplified secondary information with a
reference voltage and for outputting a second signal
indicating that a level of said amplified secondary
information is above a noise level; and

a data gate coupled to said comparator for passing
said second signal.

18.   The apparatus of claim 13, wherein said
plurality of samples is encoded by amplitude modulation.

19.   An apparatus for decoding secondary
information within a television signal, said television
signal having a first plurality of lines, said apparatus
comprising:

a terminal for receiving said television signal
containing said secondary information;

        means for selecting a second plurality of lines
among said first plurality of lines;

        means for extracting said plurality of samples from
said second plurality of lines; and

        means for decompressing said plurality of samples.

        20.   The apparatus of claim 22, wherein said means
for decompressing said plurality of samples includes
means for outputting said plurality of samples at
substantially equally spaced time intervals;

        wherein each of said second plurality of lines has
an active video region, and each of said active video
regions includes an overscan portion;

        wherein said plurality of samples are extracted
from said overscan portions of said second plurality of
lines; and

        wherein said second plurality of lines includes
only usable lines and not unusable lines.

        21.   The apparatus of claim 23 further comprising:

        a first circuit for providing timing and control
signals;

        a synchronization separation and DC restoration
circuit coupled to said terminal and said first circuit
for outputting a composite synchronization signal to
said first circuit and a clamped video signal;

        a sample and hold gate controlled by said first
circuit for passing a sample of said clamped video
signal;

        a sample and hold circuit coupled to said sample
and hold gate for holding said sample;

        an analog-to-digital converter coupled to said
sample and hold circuit for converting said sample into

a digital sample, said analog-to-digital converter
controlled by said first circuit;

a memory coupled to said analog-to-digital
converter for storing said digital sample, said memory
controlled by said first circuit;

a digital-to-analog converter coupled to said
memory for converting said digital sample into an analog
sample, said digital-to-analog converter controlled by
said first circuit;

a second circuit coupled to said first circuit and
to said sample and hold circuit for determining whether
said secondary information is valid;

a third circuit coupled to said first circuit and
to said sample and hold circuit for determining whether
a level of said secondary information is above a noise
level;

an AND gate coupled to said second and third
circuits for receiving output signals from said second
and third circuits; and

an audio gate coupled to said digital-to-analog
converter for passing said analog sample, said audio
gate controlled by said AND gate,

wherein said means for selecting said second
plurality of lines among said first plurality of lines
includes said first circuit;

wherein said means for extracting said
plurality of samples from said second plurality of lines
includes said first circuit, said synchronization
separation and DC restoration circuit, said sample and
hold gate and said sample and hold circuit; and

wherein said means for decompressing said
plurality of samples includes said first circuit, said
analog-to-digital and digital-to-analog converters, said
memory and said audio gate.

22.   The apparatus of claim 13, wherein the
secondary information is audio information.

23.   The apparatus of claim 13, wherein the
secondary information includes control information.

24.   The apparatus of claim 23, wherein the control
information identifies valid portions of secondary
information.

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FIG. I

FIG. 2
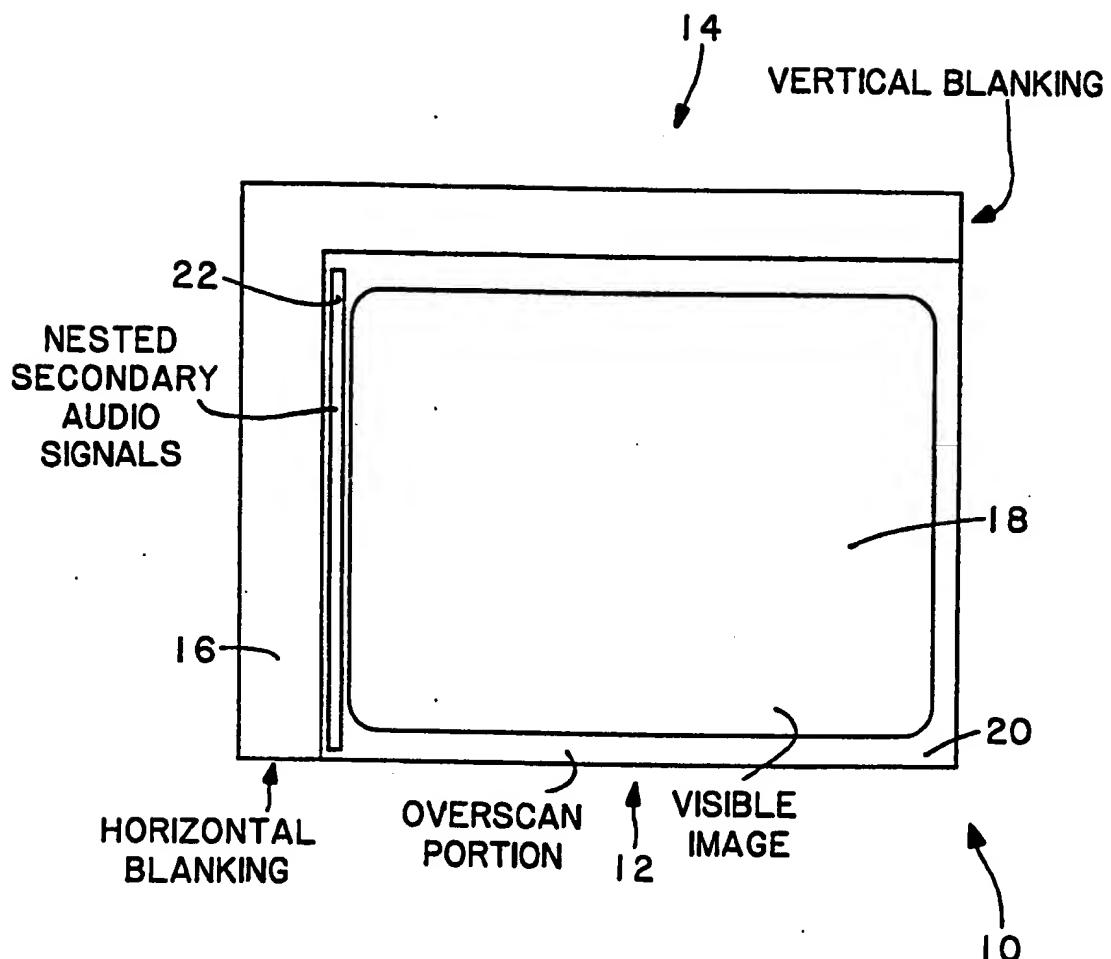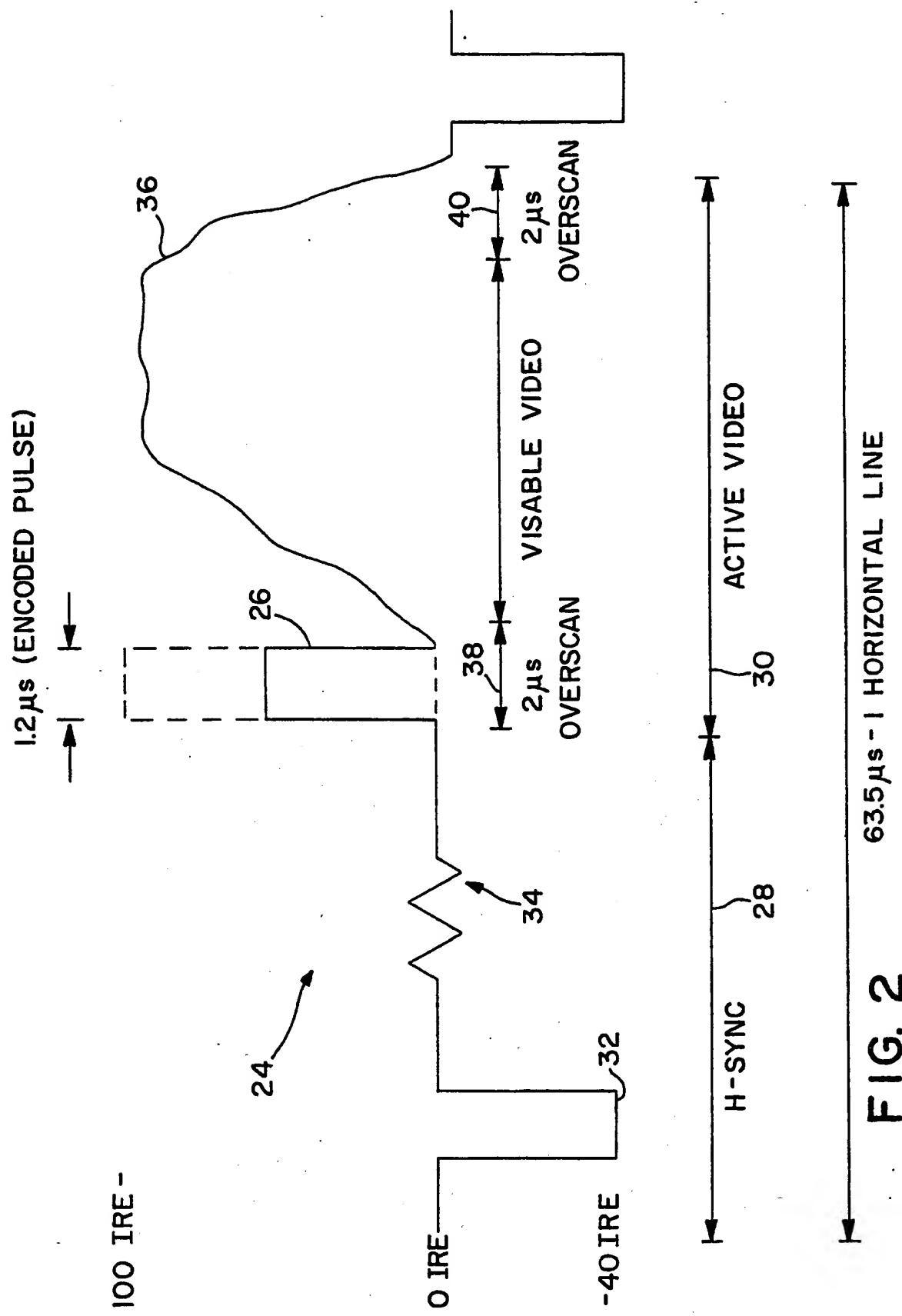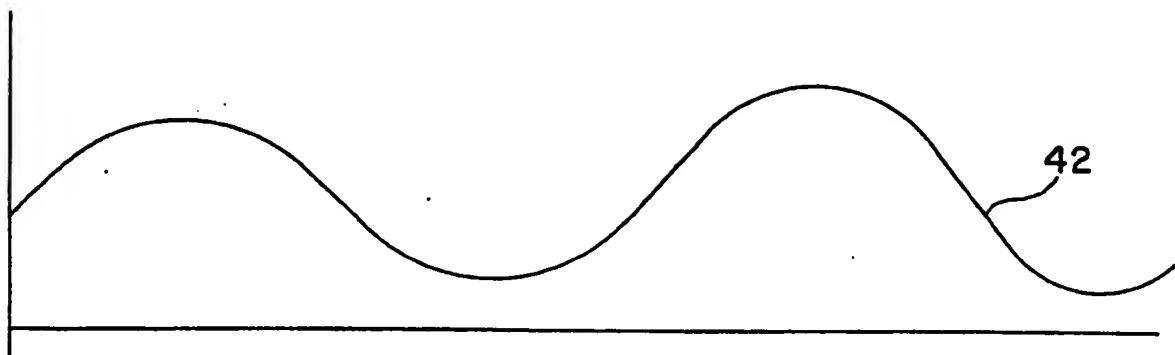
FIG. 3A

FIG. 3B

FIG. 3C

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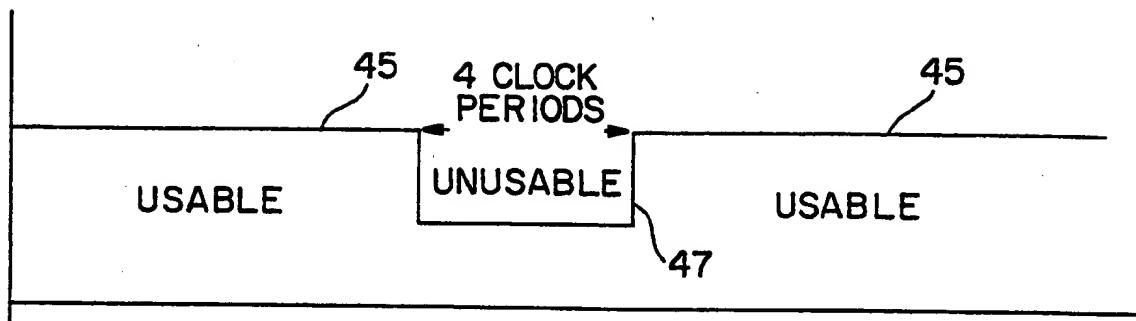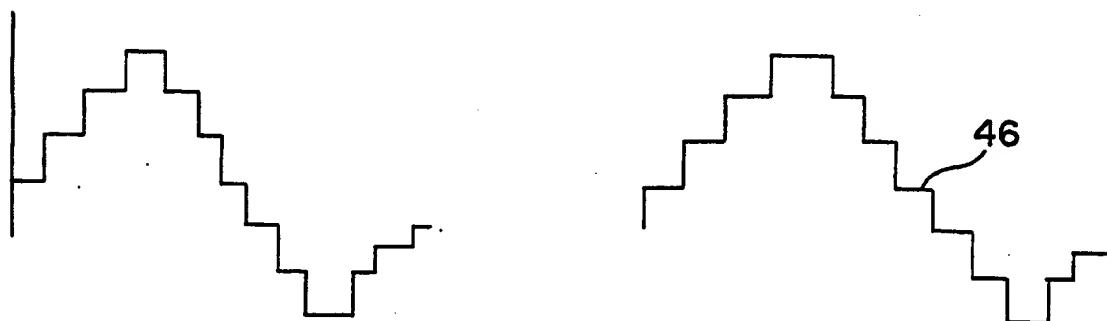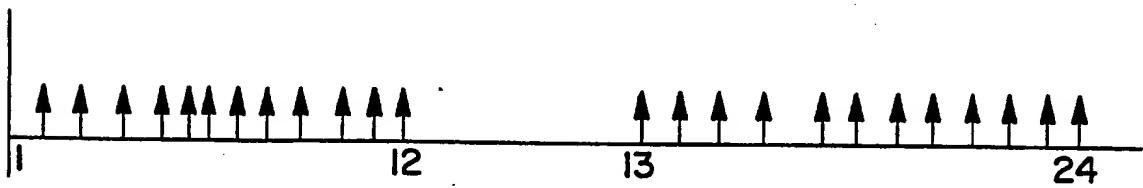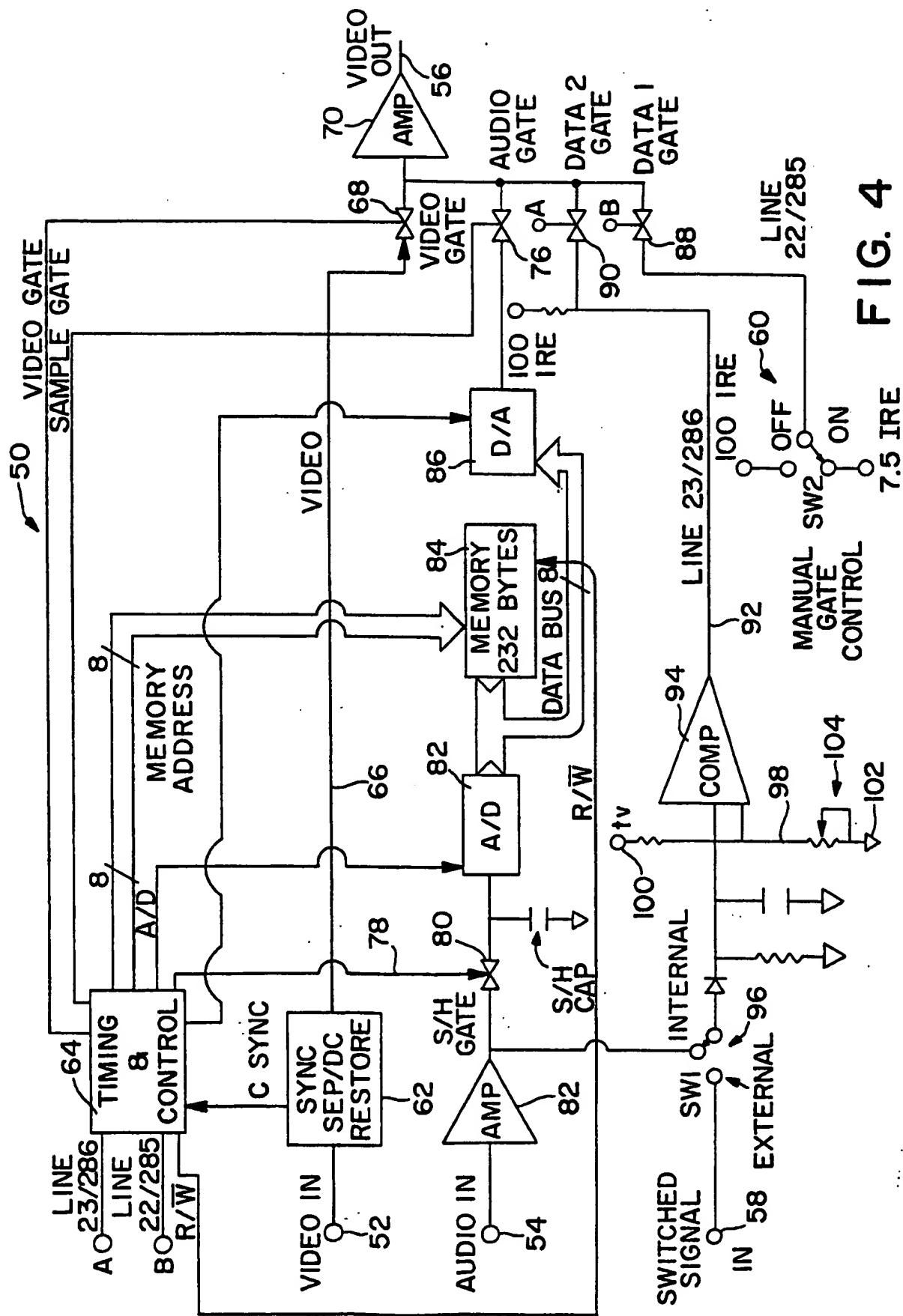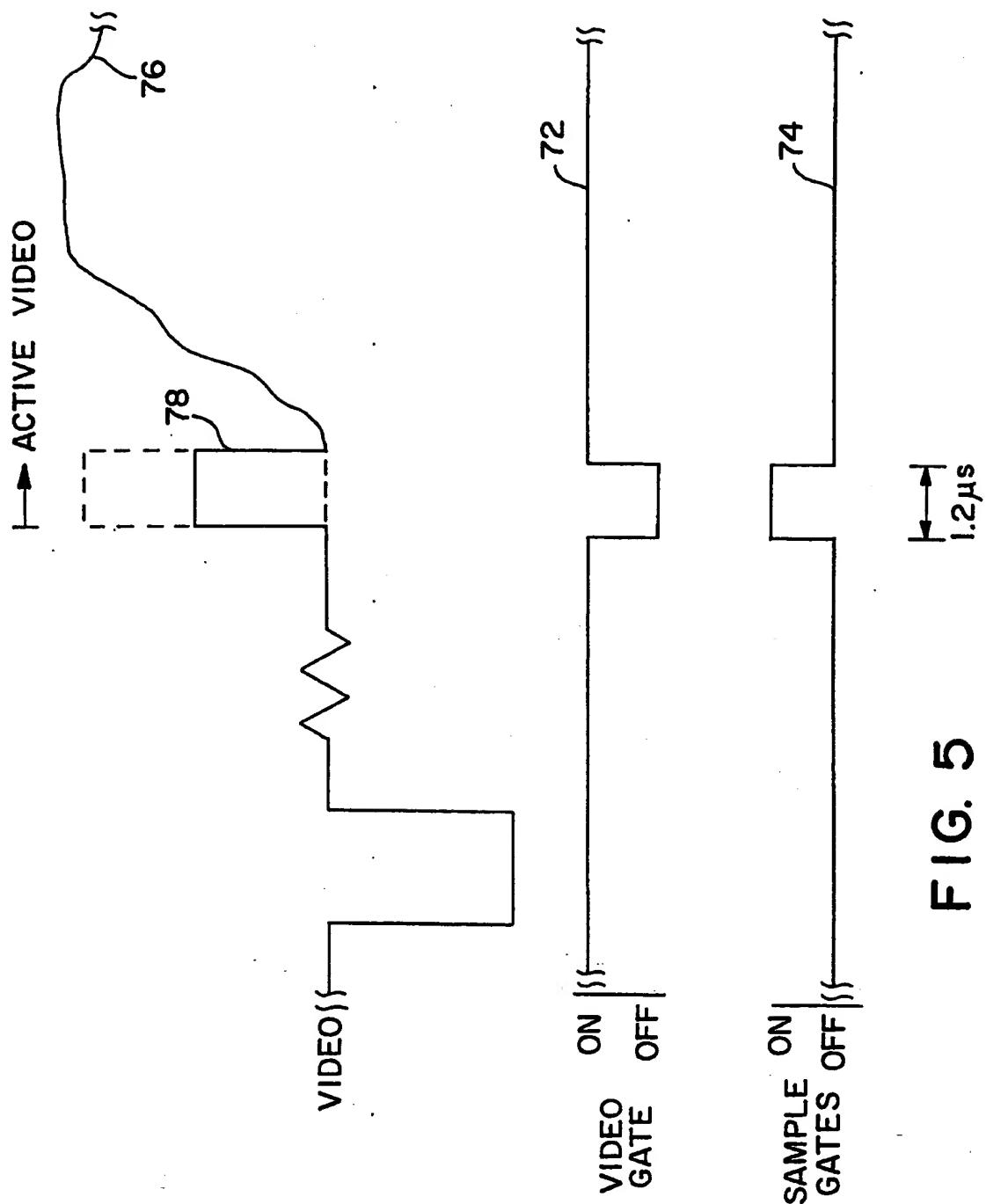
45   4 CLOCK
            PERIODS

USABLE   UNUSABLE   USABLE

                        47

FIG. 3D

46

FIG. 3E

1   12   13   24

FIG. 3F

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FIG. 4

FIG. 5
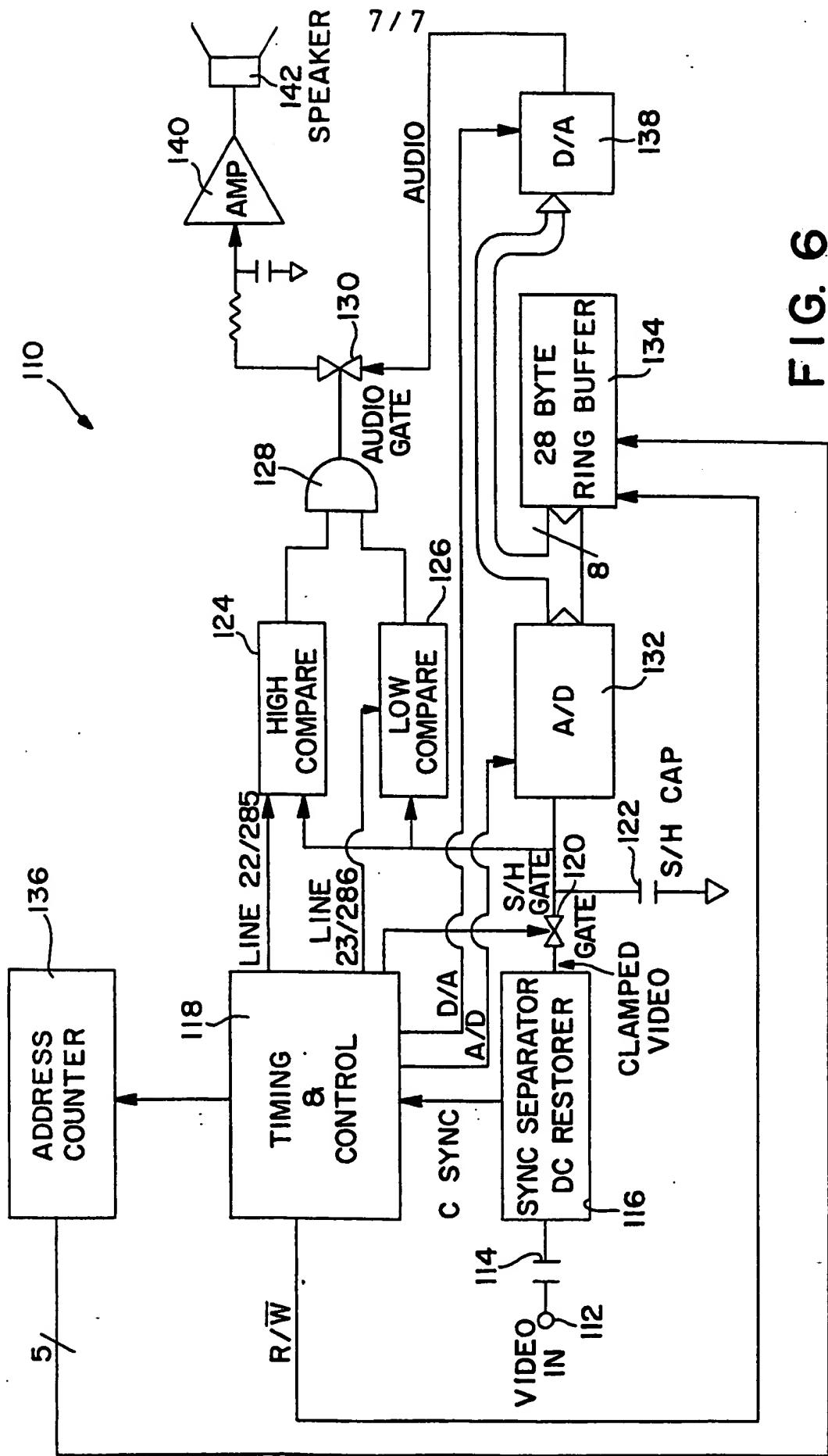
FIG. 6

	International application No.
	PCT/US95/04848

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04N 7/04, 7/025, 7/00

US CL :348/482, 476'

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 348/480-485, 476, 473, 460-462, 434; IPC(6): H04N 7/08, 7/093, 7/04, 7/025, 7/00

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS: overscan(4a)(audio or aural)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US, A, 4,665,431 (COOPER) 12 May 1987, see FIG. 6, col. 1, line 57 through column 2, line 14, and column 2, line 58 through col. 8, line 26.	1 - 9 , 1 1 , 1 3 , 14,18,22-24
X — Y	US, A, 4,295,223 (SHUTTERLY) 13 October 1981, see the whole document.	1-15,18,22- 24 — 16,17,19-21

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
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"E"	earlier document published on or after the international filing date	"X"	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"Y"	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
17 MAY 1995	28 AUG 1995

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	SAFET METJAHIC
Facsimile No. (703) 305-3230	Telephone No. (703) 305-4741
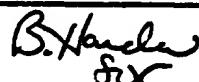
Form PCT/ISA/210 (second sheet)(July 1992)*